REMARKS

Applicant would like to thank the Examiner for the interview with Applicant's representative on October 2, 2003. As agreed to by applicant and the examiner, one aspect of the present invention is that the adaptive filters perform filtering and update the adaptive filter weight vectors in the frequency domain, whereas the references in the last Office Action filtered and updated the filter vectors in the time domain.

Claims 1-10 have been canceled, and claims 11- 33 have been added. Favorable reconsideration of the application is requested in view of the comments herein.

Claims 14-15, 19-33 are patentable over Marash (U.S. Patent No. 5,825,898) in view of Coker (U.S. Patent No. 4,581,758).

None of the references cited teach, or suggest Fast Fourier Transform means to transform the successive blocks of data microphone output signals to a frequency domain representation to facilitate filtering in the frequency domain as recited in the claims. Furthermore, the references do not teach, or suggest filtering a block of data in the frequency domain, transforming filter weight values back to the time domain using an inverse Fast Fourier transform, zeroing out portions of the filter wait values that give rise to unwanted circular convolution, and converting the filter values back to the frequency domain as recited in claims 14-15 and 19-33.

Marash shows a frequency selective constraint adaptive filter that includes a Finite Impulse Response (FIR) filter and a Least Mean Square (LMS) weight updating unit. A flat frequency reference channel passes through the FIR filter to produce a canceling signal. A difference unit subtracts the canceling signal from the main channel to generate an output signal. However, unlike the present invention, Marash does not teach that the signal is transformed to the frequency domain for comparison or for updating the filter weights, nor does Marash teach that the signal filtered in the frequency domain as recited in the claims.

Furthermore, Marash teaches a frequency selective weight constraint unit, which after the filter weights have been adapted to minimize the error, performs an FFT of the

weights, truncates the values of the filter weight representation, and converts the truncated frequency representation values with an IFFT back to new filter weight values.

In contrast, claims 11-33 update the filter weights in the frequency domain, whereas Marash does not perform the FFT until after the filter weights have already been updated. Furthermore, the present invention transforms the filter weight values back to the time domain using an inverse Fast Fourier Transform, whereas Marash converts truncated frequency representation values using an inverse Fast Fourier Transform. However, Marash stops here and stores the filter values in the time domain (see also FIG. 11, steps 400-430) and does not zero out portions of the weight values that give rise to unwanted circular convolution, nor does Marash convert the filter values back to the frequency domain.

None of the aforementioned deficiencies noted for Marash are cured by Coker. Therefore, for the reasons set forth Applicant requests these claims be considered in condition for allowance.

Claims 11-20 are patentable
over Marash (U.S. Patent No. 5,825,898) in view of Coker (U.S. Patent No. 4,581,758).

None of the references teach or show a plurality of microphones positioned to detect speech from a single speech source and noise from a noise source, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone and the data microphones receives acoustic signals both from the speech source and from the noise source, nor do they show combining the adaptively filtered output signals from the microphones in a signal summation circuit, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio as recited in the claims.

Marash shows a system wherein the outputs of a sampling unit are connected to a main channel matrix producing a main channel representing signals received in the direction of a source and contains both a source signal component and interference

signal component (col. 4, lines 60-63), The outputs of the sampling unit are also connected to a reference channel matrix unit which generates reference signals received from directions other than that of the source signal, and represent interference signals (col. 4, lines 65-68; *see also* col 3, lines 53-54 and claims 1,5 and 9). The reference signals are filtered and canceling signals are generated from the reference signals (col. 5, lines 1-7). The canceling signals are then subtracted from the main channel matrix signal by a difference unit (col. 5, lines 10-11).

Unlike Marash, as recited in the claims, each microphone is positioned to detect both signal and noise, hence the output of each data microphone as well as the reference microphone contain both data and noise (*see for example Fig. 5)*, whereas the reference microphones in Marash generate signals received from directions other than that of the source signal. Furthermore, the present invention combines the filtered signals whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio. By contrast, Marash does not combine signal components. Marash subtracts the reference signals, which contain only interfering signals from the main channel matrix. Only the main channel matrix contains both source signals and interference signals. Therefore, Marash teaches away from the present invention.

To further illustrate the difference between Marash and claims 11-20, the output after the summer in Marash equals $S_{MC} + N_{MC} - \Sigma N_{\text{ref channels}}$. The magnitude of the signal component does not change after summation because the only signal component is in the matrix channel. In contrast, claims 11-20 combine the signal component from the reference microphone and the data microphones. The output after summation = $S_{\text{ref mic}} + N_{\text{ref mic}} + \Sigma S_{\text{data mic}} + \Sigma N_{\text{data mic}}$ or $nS + N_{ref} + \Sigma N_{data}$.

None of the aforementioned deficiencies noted for Marash are cured by Coker. Therefore, for the reasons set forth Applicant requests these claims be considered in condition for allowance.

CONCLUSION

In view of the foregoing remarks, Applicant respectfully submits that the present application is in condition for allowance. Applicant respectfully requests reconsideration of this application and that the application be passed to issue.

A check in the amount of $910.00 is enclosed. Please charge any additional fees and credit any over payment to Deposit Account number 20-0090.

Respectfully submitted,

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